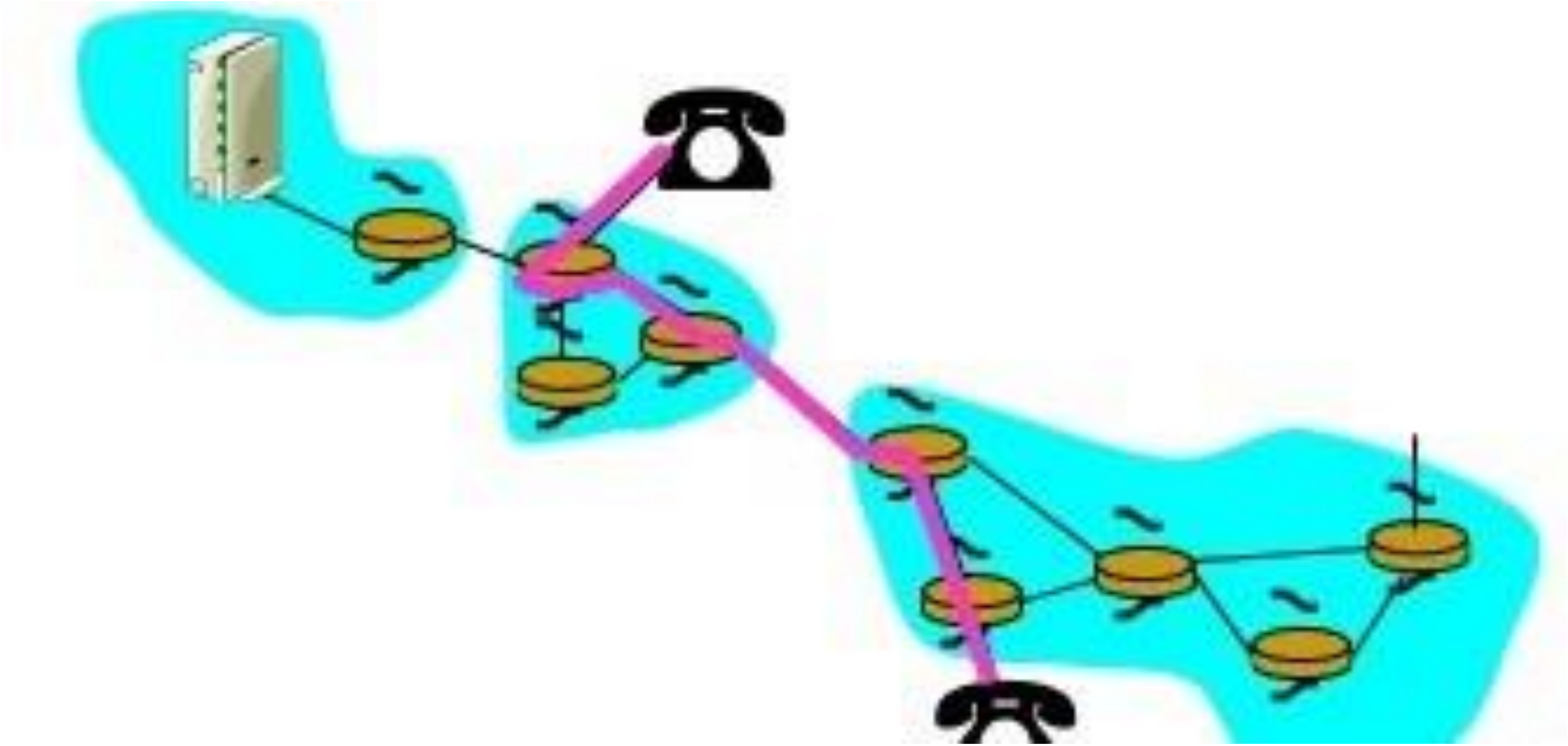
# Introduction

- Multimedia Technology that enables humans to use computers capable of processing textual data, audio and video, still pictures, and animation.
- Today, people not only use the internet to watch movies but also to upload videos (YouTube), make internet calls (Skype and Google talk).
- By the end of the decade and with emerging technologies like 4G and Wi-Fi access, Internet will not only provide phone service for less money, but will also provide numerous value-added services, such as video conferencing, online directory services, and voice messaging.

# Multimedia Networking



# Multimedia Networking Applications

- Multimedia network application is any network application that employs audio or video.
- To understand internet multimedia applications, first we look at the characteristics (properties) of video and audio.

## Properties of Video

- Video has high bit rate (100Kbps for low-quality video conferencing to 3Mps HD Movies).
- Video can be compressed. Compression can be used to create different versions of a video with each having different video quality so users can choose whichever version they can watch according to their available bandwidth.

# Properties of Audio

Basic encoding technique of converting analog into digital audio is called pulse code modulation (PCM).

Examples of Sampling Rates

- Speech encoding (uses PCM) ;8000 samples per second and 8 bits per sample.
- Audio compact disk (also uses PCM) ; 44,100 samples per second and 16 bits per sample.

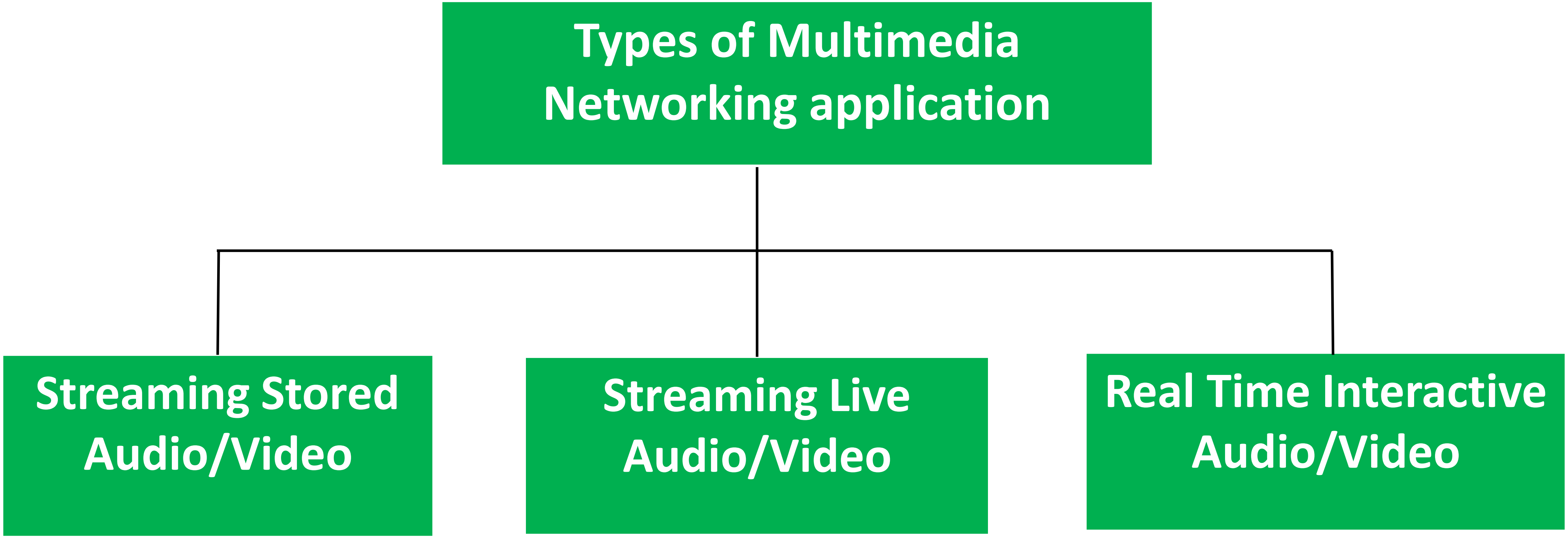
PCM encoded speech not commonly used on the internet so compression techniques are used to lower bit rates of audio streams.

- MP3 is a compression technique and the encoders can compress rates most commonly to 128 kbps.

Note that digital audio has lower bandwidth compared to video although users are more sensitive to audio malfunctions than video.

# Types of Multimedia Networking application

Multimedia networking applications are classified into 3 categories.



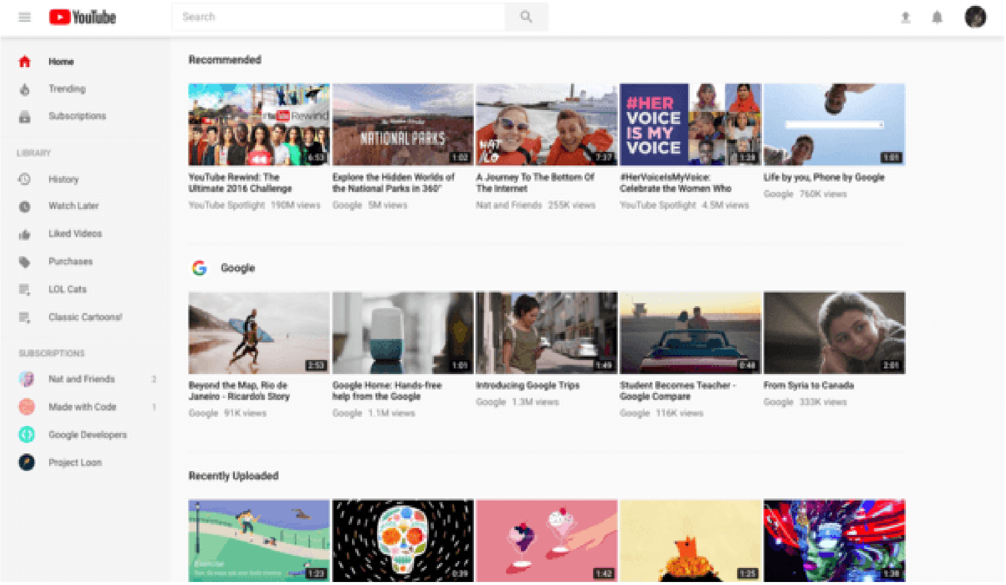
# Streaming Stored Audio/Video

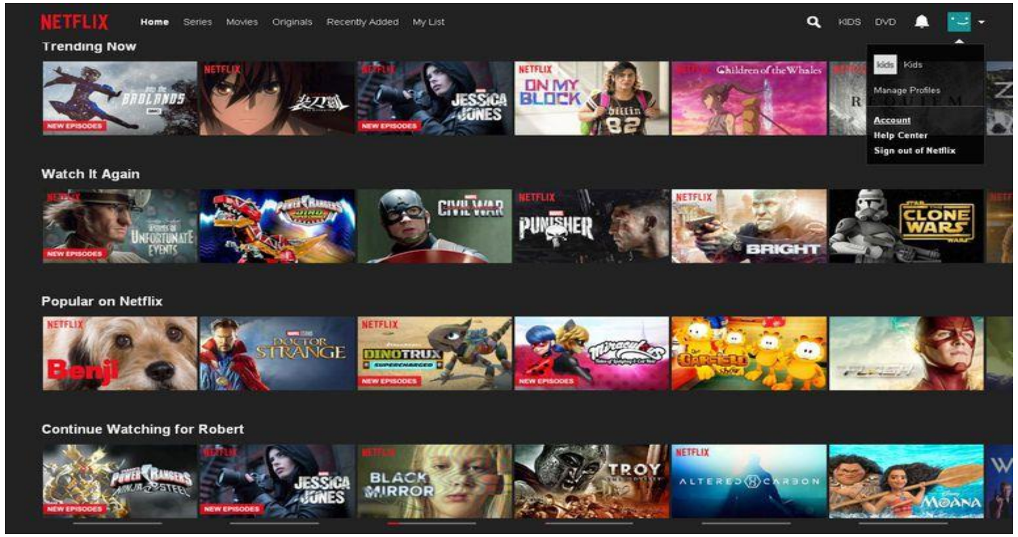
➢ **Stored:**

- Prerecorded audio/video are stored at server.
- Users send requests to the servers to view the videos.
- Users may fast rewind, pause, fast forward the multimedia content.
- Response time should be in the order of 1-10 seconds.

➢ **Streaming:**

- Simultaneous play out and download.



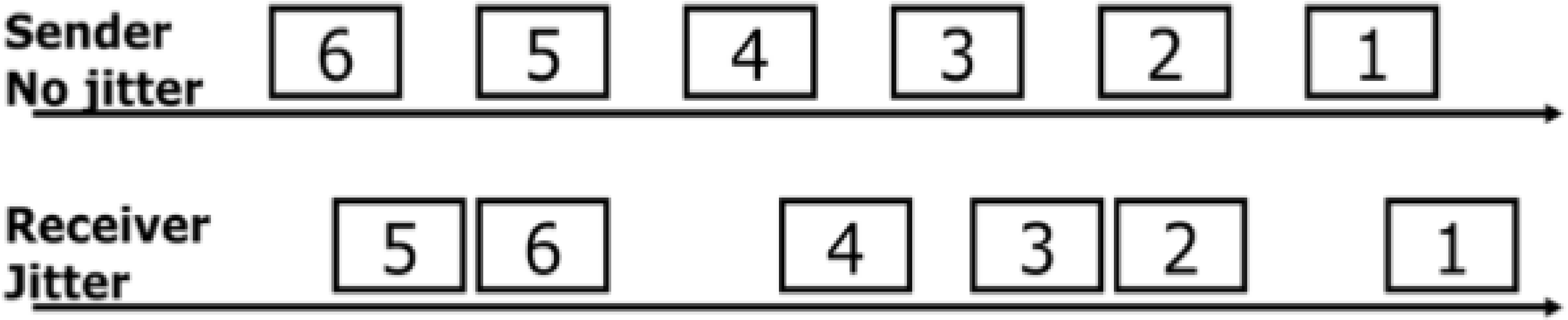



# Streaming Stored Audio/Video(cont.)

➢**Continuous playout:**

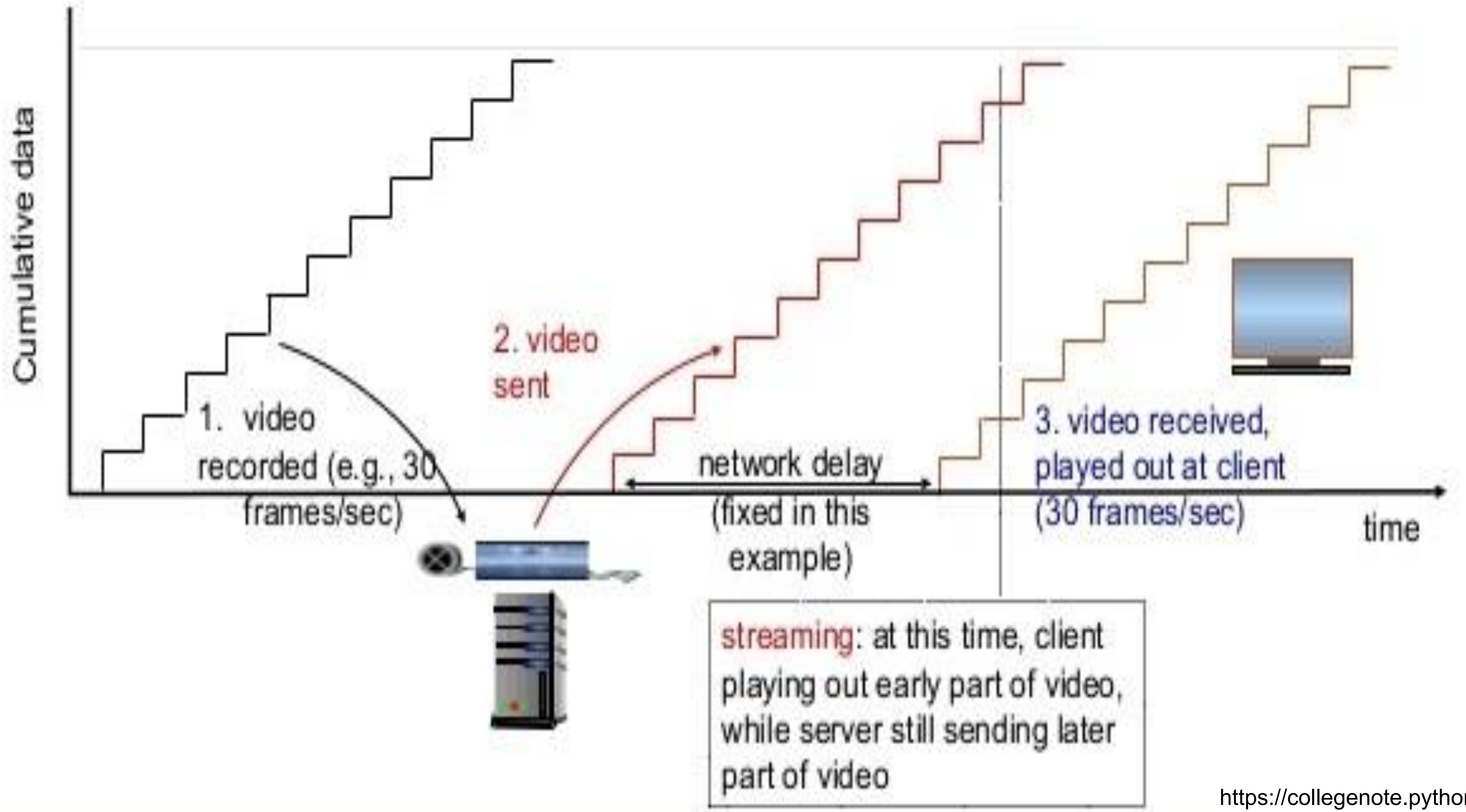
- Once play out begins, it should proceed based on the original timing of the recording.
- Delay jitter smoothed by playout buffer.

Jitter is the variability of packet delays within the same packet stream.




**Examples:** Youtube, Netfilx, etc.

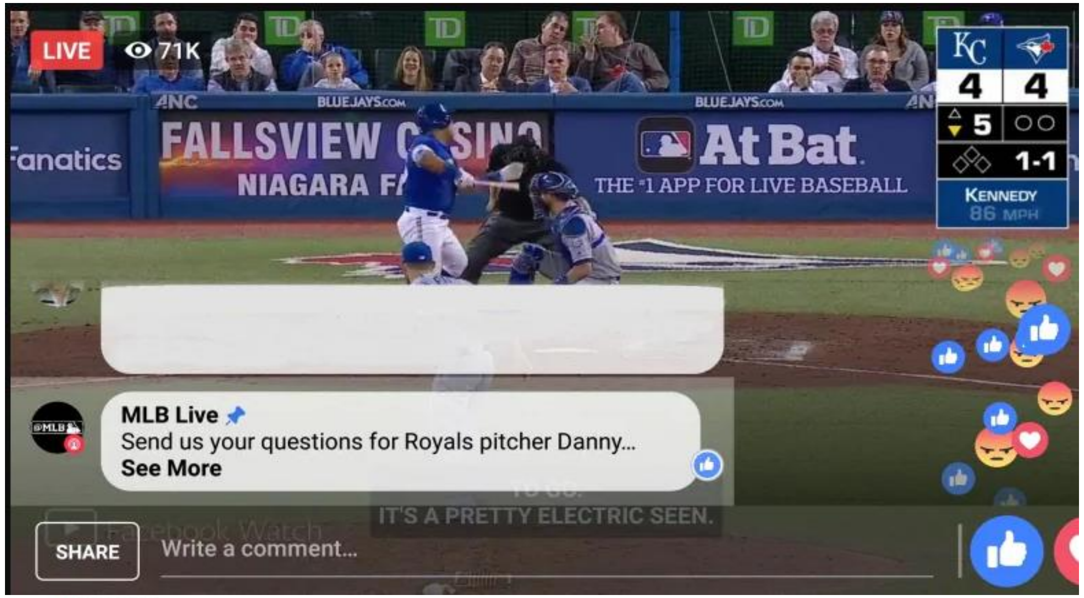
# Streaming Stored Video

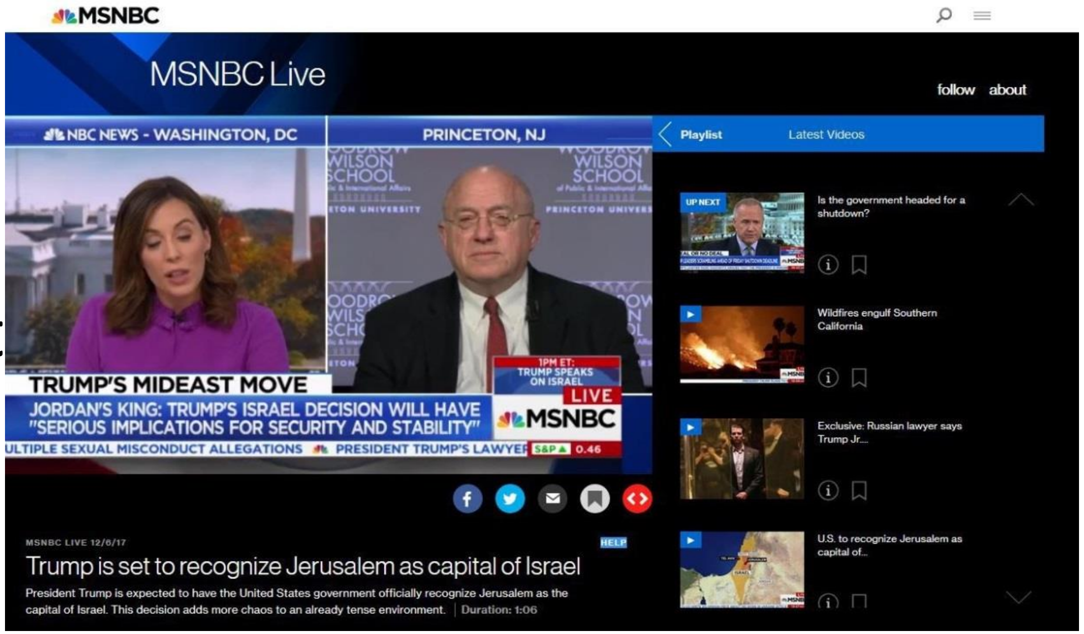


# Streaming Live Audio/Video

- Similar to traditional radio and television, except the audio/video contents are transmitted on the internet.
- Fast forward impossible.
- Rewind and pause possible.
- High data rate to large number of users.
- Requires continuous play out and high quality on the end-to-end delay.
- Delay jitter controlled by caching.

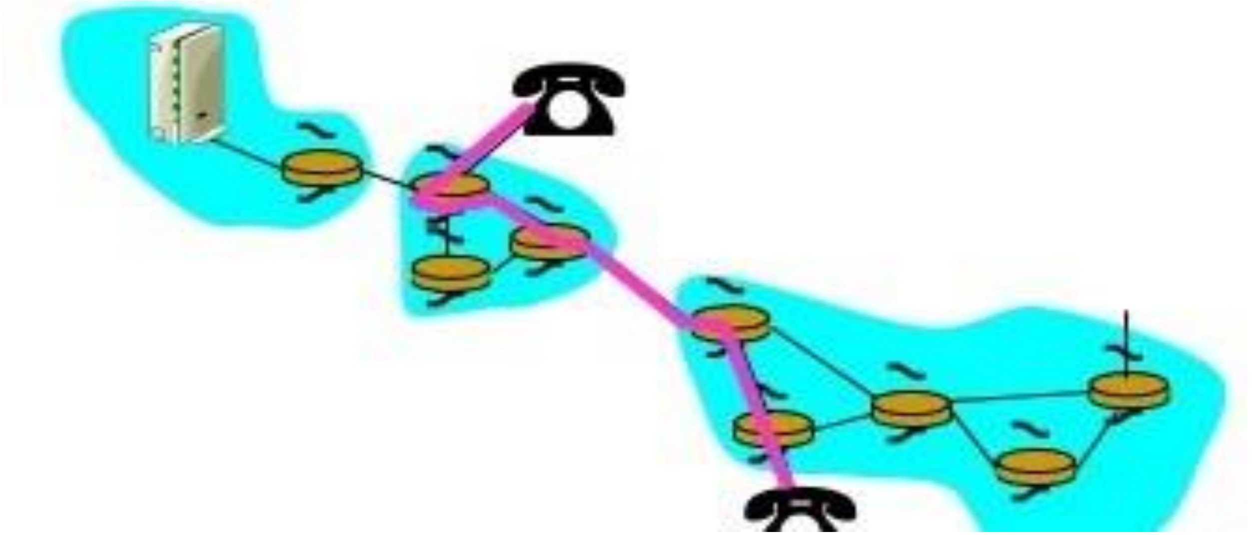
**Examples**: Live sporting events, live internet news feed, internet radio talk show.

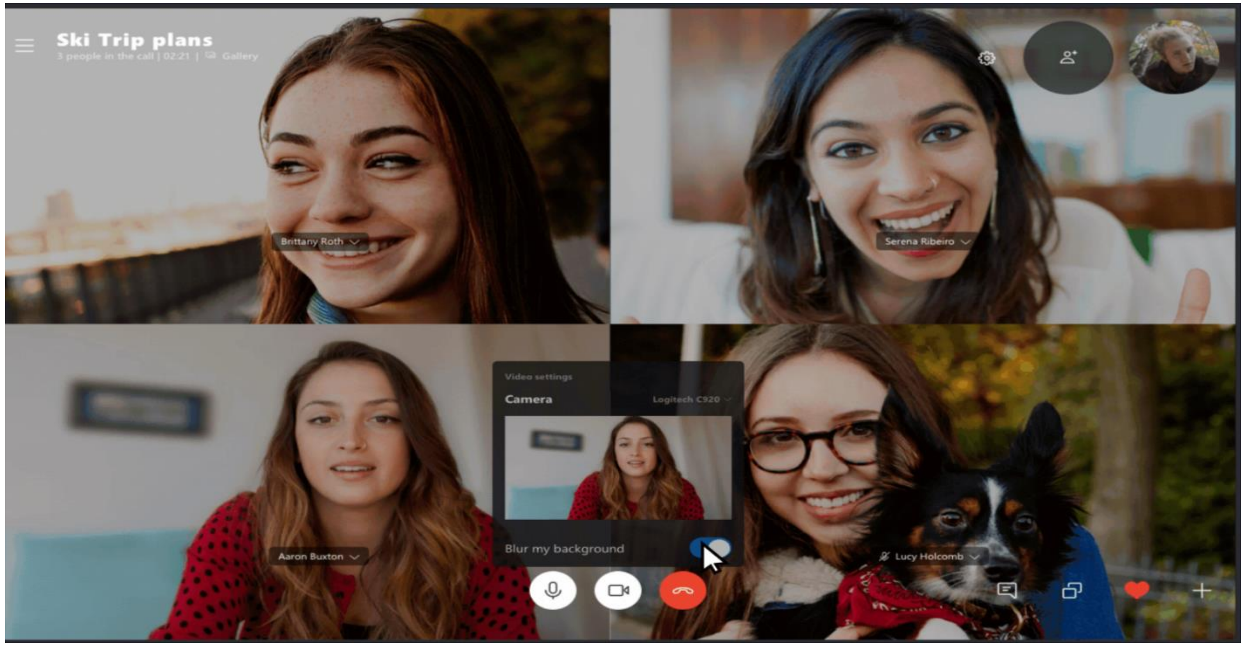


# Real Time Interactive Audio/Video

- Also called conversational voice/ video over IP.
- Allows users using audio/video to communicate with each other in real time.
- Requires very high quality on the end-to-end delay, usually a fraction of one second.
- Session initialization





# Real Time Interactive Audio/Video

**Examples**: Internet Telephone,
Video Conferencing,
Skype, Online games,
etc.

# Session Initiation Protocol (SIP)

Session Initiation Protocol (SIP) is a signaling protocol used for initiating, maintaining, modifying and terminating real-time sessions that involve video, voice, messaging and other communications applications and services between two or more endpoints on IP networks.

SIP was developed by the Internet Engineering Task Force (IETF) to address the evolving needs of IP-based communications. Native support for mobility, interoperability and multimedia were among the drivers behind SIP's development. SIP complements other communications protocols, such as Real-Time Transport Protocol (RTP) and Real-Time Streaming Protocols (RTSP), used in IP-based sessions.

# SIP Network Elements

There are some entities that help SIP in creating its network. In SIP, every network element is identified by a **SIP URI** (Uniform Resource Identifier) which is like an address. Following are the network elements –

- User Agent
- Proxy Server
- Registrar Server
- Redirect Server
- Location Server

# How SIP Works?

- Like HTTP or SMTP, SIP works in the application layer of the Open Systems Interconnection (OSI) communications model.
- SIP is a request-response protocol, receiving requests from clients and responses from servers. Requests can be sent through any transport protocol, such as UDP, SCTP or TCP.
- SIP determines the end system to be used for the session, the communication media and media parameters, and whether the called party agrees to engage in communication. Once these are assured, SIP establishes call parameters at either end of the communication, also handling call transfer and termination

# H.323

- ➢ A popular standard for real-time audio and video conferencing among end systems in the Internet.

The standard includes the following:

- ✓A specification for how endpoints negotiate common audio/video encodings.
- ✓ It mandates RTP for audio and video data encapsulation and transmission over the network.
- ✓A specification for how endpoints communicate with their respective gatekeepers (a device similar to and SIP registrar).
- ✓A specification for how Internet phones communicate through a gateway with ordinary phones in the public circuit-switched telephone networks.

# Resource Reservation Protocol (RSVP)

- ✓RSVP is a protocol used by sessions to send the reservation request and does not specify how the network provides the reserved resources.
- ✓It is not a routing protocol either and does not determine the links in which the reservations are to be made.
- ✓RSVP is receiver-oriented. The receiver of a data flow initiates and maintains the resource reservation.
- ✓It is used by application sessions to reserve resources in the Internet.
- ✓It is used to reserve bandwidth for multicast trees (unicast is treated as a degenerate case of multicast).

# RSVP Reservation Type

1. Distinct Reservation: Receiver request to reserve operation of bandwidth for each sender.
2. Shared Reservation: To reserve common resources for all sources.

# Working:

✓RSVP operates in a two-pass_manner.

✓A transmitting source advertises its content by sending an RSVP path message through a multicast tree, indicating the bandwidth required for the content, the timeout interval, and information about the upstream path to the source.

✓ Each receiver sends an RSVP reservation message upstream on the multicast tree.

✓ The reservation message specifies the rate at which the receiver wants to receive the data.

✓ When a router receives a reservation message, it first checks if its downstream links can accommodate the reservation. If yes, it adjusts its packet scheduler to accommodate the reservation and sends a reservation upstream on the multicast tree. Otherwise it rejects the reservation and sends an error message to the corresponding receivers.

# MULTIMEDIA NETWORKING PROTOCOLS

## RTP(Real Time Protocol)

- RTP is a network protocol for delivering audio and video over IP networks.
- This protocol is used for real time data transport,in this case video and audio.

# RTP packet header fields include

| | | | | |
|---|---|---|---|---|
| **Payload Type** | **Sequence number** | **Timestamp** | **Synchronization Source Identifier** | **Miscellaneous fields** |

- Payload type ,7 bits used to indicate the type of encoding for audio and video.
- Sequence number,16 bits,incremented by one for each RTP packet sent;
- Timestamp ,32 bits,used to give the sampling instant of the first byte in the RTP packet data;
- Synchronization source identifier(SSRC),32 bits,used to identify the source of the RTP stream;
- Miscellaneous fields

# RTCP(Real Time Control Protocol)

- RTP is a protocol that provides basic transport layer for real time applications but does not provide any mechanism for error and flow control,congestion control,quality feedback and synchronization.
- For that purpose the RTCP is added as a companion to RTP to provide end-to-end monitoring and data delivery ,QoS(Quality of Service)

# RTCP is responsible for three main functions:

- Feedback on performance of the application and the network.
- Correlation and synchronization of different media streams generated by the same sender(eg;combined audio and video)
- The way to convey the identity of sender for display on a user interface.